REMARKS

The claims have been amended to more clearly define the invention as disclosed in the written description. In particular, claims 15 and 16 have been cancelled.

Applicants believe that the above changes answer the Examiner's objection to claim 15 and respectfully request withdrawal thereof.

The Examiner has rejected claims 1, 2, 4, 6-9 and 11-16 under 35 U.S.C. 13(a) as being unpatentable over U.S.P.A.P. No. 2002/0154041 to Suzuki et al. The Examiner has further rejected claims 3 and 5 under 35 U.S.C. 103(a) as being unpatentable over Suzuki et al. in view of U.S. Patent 7,110,554 to Brennan et al. In addition, the Examiner has rejected claim 10 under 35 U.S.C. 103(a) as being unpatentable over Suzuki et al. in view of U.S. Patent 5,740,523 to Nakajima et al.

The Suzuki et al. publication discloses a coding device and method, decoding device and method, and recording medium, in which an adaptive mixing section performs a mixing process on input signals on the basis of distortion factor information supplied from a distortion factor detection section, and controls the operation time of MS stereo coding or IS stereo coding, and creates power correction information in accordance with a mixing coefficient and causes power correction to be performed during reproduction.

The Examiner has indicated that Suzuki et al. includes all of the steps claimed in claim 1. In particular, the Examiner indicates that the limitation "dividing said at least two input

audio signals into a plurality of sequential segments" is disclosed in Suzuki et al. at page 2, paragraph [0018] with page 5, paragraph [0085].

Applicants submit that the Examiner is mistaken. In particular, paragraph [0018] (paragraph [0085] presents similar limitations) states:

> "A filter bank 31-1 divides an input left signal L(t) into signals $L_n(t)$, $L_{n-1}(t)$, . . . , $L_1(t)$ (n is the number of divided bands) of predetermined frequency bands, and outputs each signal to a corresponding dual coding section 32 and a corresponding MS/IS coding section 33. In FIG. 3, although only the dual coding section 32 and the MS/IS coding section 33 for processing the signal $L_n(t)$ are shown, coding sections corresponding to signals $L_{n-1}(t)$, . . . , $L_1(t)$ are provided in a similar manner." (paragraph [0019] provides similar processing for an input right signal R(t).)

It should be apparent from reading the above that $L_n(t)$, $L_{n-1}(t)$, . . . , $L_1(t)$ are not sequential segments but rather are in parallel. This can be seen graphically in Fig. 3 where the filter bank 31-1 has multiple outputs corresponding to $L_n(t)$, $L_{n-1}(t)$, . . . , $L_1(t)$. If the band signals of Suzuki et al. were sequential, there would only need to be one output.

The Examiner further indicates that the limitation "calculating, for each of the sequential segments, a correction factor for each of a plurality of frequency bands (i) as function of the energy of the frequency components of the summed frequency components in said band ($\sum_{k\in i}|S(k)|^2$) and the energy of said frequency components of the input audio signals in said band

$(\sum_{k \in i} \{ |L(k)|^2 + |R(k)|^2 \})$" is disclosed in Suzuki et al. at page 5, paragraph [0093].

Again, Applicants submit that the Examiner is mistaken. In particular, paragraph [0093] states:

> "A power computing section 121 computes the power values Pl_n and Pr_n from the signals $L_n(t)$ and $R_n(t)$ which are divided into predetermined bands by the filter banks 101-1 and 101-2, respectively, and outputs them to a power correction section 123."

It should be apparent from the above that Suzuki et al. only uses the power of the input audio signals in each band and not the energy of the frequency components of the summed frequency components. This should be clear from Fig. 9 in which for each frequency band, a power computing section 121 computes the power information and a power correction section generates that power correction information which is sent by the adaptive mixing section 102 to the multiplexer 107.

In addition, the Examiner indicates that the claim limitation "correcting each summed frequency component as a function of the correction factor (m(i)) for the frequency band of said component" is disclosed in Suzuki et al. at page 5, paragraph [0087].

Applicants submit that the Examiner is again mistaken. In particular, paragraph [0087] states:

> "Furthermore, the adaptive mixing section 102 creates power correction information $P_{n,adj}(t)$ for correcting the output of the left and right signals, and outputs it to a multiplexer 107."

However, the functioning of the multiplexer 107 is described at paragraph [0091] which states:

> "The multiplexer 107 combines a code sequence C_n of a predetermined band, supplied from the coding section 105 with the code sequences C_{n-1}, . . . , C_1 of the other bands, and outputs the combined audio data C to a device (not shown) provided external to a coding device 91, a network, etc. The combined audio data C contains power correction information $P_{n,adj}(t)$ supplied from the adaptive mixing section 102 and information indicating by which coding method the signals are coded."

It should be clear that Suzuki et al. does not correct the code sequence C_n using the power correction information (which, as indicated above, is not equivalent to the correction factor as claimed), but rather merely combines this power correction information with the code sequence C_n to form the output signal C.

The Brennan et al. patent discloses sub-band adaptive signal processing in an oversampled filterbank.

The Examiner has indicated that Brennan et al. discloses the claim 3 limitation "combining, for each input audio signal, overlapping segments into respective time-domain signals representing each input audio signal for a time window".

Applicants submit that the Examiner is mis-reading Brennan et al. In particular, while Brennan et al. discloses a filterbank for filtering an input signal into a plurality of overlapping frequency bands, there is no disclosure of respective time-domain signals "representing each input audio signal for a time window". Further, Brennan et al. does not supply that which is missing from Suzuki et al. as noted above.

The Nakajima et al. patent discloses a radio receiver which discloses the use of "linear interpolation" to determine SPC (Separation Control) and HCC (High Cut Control) voltages to compensate for variations in the characteristics of various FM multiplexer IC's.

The Examiner now states that Nakajima "discloses a method wherein the correction factor for each frequency component is derived from a linear interpolation of the correction factor for at least one band (linear interpolation; column 9, lines 55-67 and column 11, line 59 - column 12, line 11), to allow easy adjustment."

Applicants have read Nakajima et al. and outside of just mentioning the term "linear interpolation", there is no disclosure of "frequency component" nor of this linear interpolation being used to determine a correction factor for this undisclosed "frequency component" from a correction factor for at least one band. Notwithstanding this, Nakajima et al. does not supply that which is missing from Suzuki et al. as noted above.

In view of the above, Applicants believe that the subject invention, as claimed, is not rendered by the prior art, and as such, is patentable thereover.

Applicants believe that this application, containing claims 1-14, is now in condition for allowance and such action is respectfully requested.

Respectfully submitted,

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